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Survey of the Acoustics of Concert Halls in European Countries

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Since we launched our digital sound processor for in-car audio two years ago, several companies have followed with their own sound processors. Customers have not, however, always been pleased with the sound; it is exaggeratedly effected to present the liveliness of a concert hall because reproducing natural sound fields is still not fully understood.
We decided to analyze the sound fields in actual buildings to produce a sound processor able to present a more accurate and lifelike sound field in the car compartment. We measured the acoustic characteristics of eight European concert halls, including some famous ones, in November 1990. This paper describes the concert halls we visited, and our measurement methods. We analyze the data we collected and talk briefly about our future plans.

1. Introduction

A car passenger compartment is not the ideal audio environment. The loudspeaker positions are asymmetric with respect to the listening position and there are sound reflectors, such as glass, and sound absorbers, such seats and passengers, in a small space. Even with an equalizer, it is very difficult to reproduce the liveness and stereo effect possible in listening rooms.

In 1989, Fujitsu Ten released their DSP sound field processor. This device made it possible to recreate the liveness of a listening room or concert hall in a passenger compartment. Since then, we have continued research and development to design products which can give listeners more natural sound. We have made it our aim to reproduce the sound field of a concert hall. However, there are many sorts of concert hall, each one having its individual shape and building materials. In November 1990, we visited four countries, Belgium, Germany, Austria, and Switzerland and recorded the acoustic characteristics of seven famous concert halls and one church. We plan to reproduce the acoustic properties of these halls.

We recorded acoustic properties in three ways:

① We measured impulse responses with closely-located 4-point microphone method.
② We measured binaural impulse responses using a dummy head.
③ We recorded music played in an anechoic room using a binaural dummy head. The music was played on audio equipment, not by musicians.

There were ten persons in our group (Figure 1): Masayuki Morimoto, an assistant professor, and Kimihiro Sakagami, an assistant in the department of technology of Kobe University; an assistant and a student of the technology department of the Ruhr University in Germany; and six persons from Fujitsu Ten. We measured the acoustic

Figure 1. Departure (at Rulr University)

characteristics of the Maximum Auditorium at Ruhr University. We recorded our first measurements here, and made a test recording with members of Ruhr University. From Ruhr university, we chartered a coach and covered more than 4000 km traveling around Europe. This paper begins with an outline of the acoustic characteristics of concert halls; it gives an analysis of our data beginning with a brief description of the concert halls we visited.

2. General

2.1 Relationship between the shape of a concert hall and its acoustic characteristics

In Japan, a number of public halls were constructed after World War II. Many of these halls are multiple-purpose and are used for a variety of events including concerts and lectures. More recently, local governments are adopting for halls designed for a particular purpose and building specialized concert halls dedicated to music. The differences between general-purpose halls and concert halls are summarized below.

The typical general-purpose hall is fan shaped, its sidewalls are further apart from the stage. This kind of hall has a shorter reverberation time than dedicated concert halls because speech must be clearly audible.

The classical concert hall is called a shoebox type because it's shaped like a shoe box. Examples of this type of hall are the Grosser Musikvereinssaal in Vienna, completed in 1870, and the Symphony Hall in Boston, completed in 1900. Since these halls are longer than they are wide, there is a large initial sound reflection is from the side walls which surround the listener. This effect is weaker in fan-shaped halls. The early reflection plays a large part in imparting a stereo effect and liveness. The shoebox shape was widely used for halls built between the end of the 19th century and the beginning of the 20th century and many people still believe that this is the best shape for concerts.

Some recent halls have been larger-capacity and of a new arena or vineyard type in which seats are arranged around the stage; the New Philharmony Hall in Berlin and Suntory Hall in Tokyo are examples. This style has the following characteristics: The stage is clearly visible from any seat. The early reflection reaches every seat because the internal shape of the walls and the relationship between the seats and sound reflecting boards suspended from the ceiling have been improved.

Concert hall shape should be designed with ideal early reflection and reverberation. Concert halls have better acoustic characteristics than general-purpose halls.

2.2 Reverberation time

The major terms used to describe the sound characteristics of a hall are presence, reverberation, and volume level.

A widely used figure is the reverberation time. The reverberation time is defined as the time from sound production to the time the sound level drops by 60 dB.

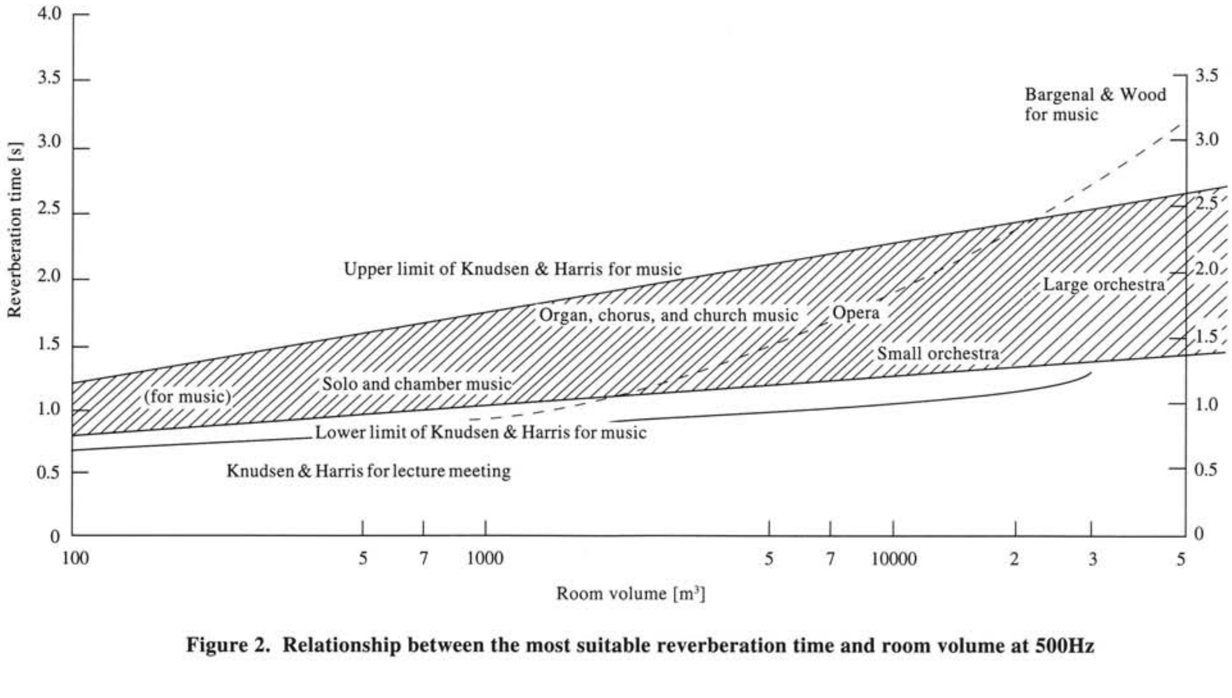


Figure 2. Relationship between the most suitable reverberation time and room volume at 500Hz

Factors influencing reverberation time include room volume, room surface area, sound absorption coefficient of the wall material, and the ratio of the audience size to the room volume. Although some people say the most suitable reverberation time for concert halls is 2 seconds, this value actually varies from one hall to another and depends on the room volume, the genre of music, and the scale of the performance. Figure 2 shows an example of the relationship between the most suitable reverberation time and the room volume at 500 Hz. The reverberation time is not, however, a comprehensive measure. For example, even if two halls have the same volume and reverberation time, they have different sound qualities depending on the timbre of reverberated sound, and the balance of direct sound and early reflection. The reverberation time does, nevertheless, give some indication of the quality of the hall.

3. Introduction to the halls whose acoustic characteristics we measured

3.1 Palais des beaux-arts, Brussels

The Palais Des Beaux-Arts of Brussels, Belgium, which was founded in 1929, is characterized by its egg shape and main floor which is divided into two parts. It has a three-story balcony along the egg-shaped sidewalls from the side of the stage to the rear of the seats. Its volume is 12,500 m^3, it has 2150 seats, and its reverberation time is about 1.4 seconds—comparatively short. According to Winckel[3)], a questionnaire in 1945 about the favorite halls of European conductors indicated that the Palais Des Beaux-Arts was as popular as the Grosser Musikvereinssaal in Vienna, the Symphony Hall in Boston, and the Concertgebouw in Amsterdam which were considered the best three halls in the world.

3.2 New Philharmony Hall

The New Philharmony Hall founded in West Berlin, Germany in 1963 is well known as a successful modern architectural acoustic design and a beautiful building. Its volume is 26,000 m^3, comparatively large; the internal shape is a typical arena or vineyard type; the layout of seats is complex because it was designed with both appearance and acoustic characteristics in mind. This hall is the home of the Berlin Philharmonic Orchestra which had both Furtwangler and Karajan as previous regular conductors. The conductor is presently Claudio Abbado and this is one of the most famous concert halls in the world.

3.3 Schauspielhaus

The Schauspielhaus was founded in 1821 in East Berlin and is famous for the concert commemorating the unification of East and West Germany, conducted by the late Leonard Bernstein and others in 1989. As its name implies, it was founded as a theater, it was burned down in World War II, then rebuilt as a concert hall in 1984. The trim is very grand — the walls, ceilings, lobbies are decorated with busts, pictures, and sculptures. The entire hall including the seats, chandeliers, and doors, are very plush. It is thought to be one of the world's most beautiful halls. This hall is characterized by its shoebox shape and has its first floor in front of the stage and a two-story balcony at the side of and behind the first-floor seats. It has 1301 seats. Although we do not have precise figures for the hall volume and other characteristics, we estimate the volume is about 10,000 m^3. Incidentally, our measurements were probably the first made for this hall.

3.4 Dom Zu Passau

Passau is a city in Southern Germany close to the border with Austria where the Danube and the Rhine meet. The Dom Zu Passau is a church which was constructed in the late 13th century and most of the building is made of stone. The room volume, including the altar placed in a secluded section, is about 110 m (depth) × 30 m (width)× 20 to 30 m (height). It is roughly twice to five times greater than that of most concert halls. The pipe organ in the church is the biggest in the world: It has 17,388 pipes, 231 stops, and 4 chimes.

3.5 Grosser Musikvereinssaal

The Grosser Musikvereinssaal was built in 1870 as a concert hall for the Vienna Philharmony Orchestra. As mentioned before, the sound quality of this hall has the reputation of being one of the three best in the world. Many fine performances have been given and recordings made here. This hall is known for its abundant gold ornamentation. There are thirty-two gold images of women, a number of sculptures, pictures, chandeliers, and other decorations. This hall has the typical shoebox shape. Its volume is 15,000 m^3, it has 1680 seats, and its reverberation time is about 3.3 seconds when the seats are vacant and about 2 seconds when the seats are filled.

3.6 Grosser Tonhallesaal

The Grosser Tonhallesaal which lies by Zurich See was completed in 1895 and remodeled in 1930. This shoebox shaped hall has a balcony along the sidewalls from the side of the stage to the rear of the seats. The main wall colors are light brown and gold. Two big chandeliers are suspended from the ceiling which has a picture frame design with a large picture on it. Its volume is 11,400 m^3 and it has 1546 seats. Although the reverberation time is long — 3.5 seconds — when the seats are vacant, it is considerably shorter — 1.6 seconds — when all seats are filled.

3.7 Stadt-Casino

Like the Grosser Tonhallesaal, the Stadt-Casino, built in 1876, is a shoebox-shaped hall. It has a balcony along the sidewalls from the side of the stage to the rear of the seats. The balcony and ceiling and pillars under the balcony are mainly white; the area under the balcony and the pillars above the balcony are mainly reddish brown. Its volume is 10,500 m^3 and it has 1400 seats. The reverberation time is about 2 seconds when the seats are vacant. This hall has long been regarded as the best in Switzerland.

4. Measurement method

4.1 Measurement with closely-located 4-point microphone method

The closely-located 4-point microphone method has been introduced in previous Fujitsu Ten technical journal. Four microphones placed close together are used to stereoscopically analyze the reflected sound which arrives at the measurement point (Figure 3). We have used this method to evaluate sound field control and to collect reference data, under the supervision of Dr.Yoshio Yamazaki, of the school of Science and Engineering of Waseda University, who developed this measurement method.

In the 4-point method, the reflected sound path is assumed as follows: In Figure 4, sound emitted from the actual sound source is reflected by the wall and reaches the measurement point (solid line). The reflected sound is regarded as a direct sound emitted by a image sound source which, like a image light source, is behind the walls (dotted line).

Given this, the intensity of the reflected sound and the direction from which the reflected sound is arriving can be seen by plotting the distribution of the image

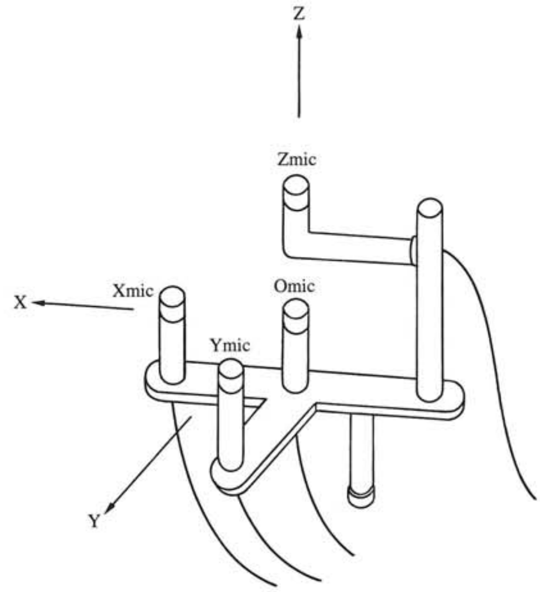


Figure 3. Microphone arrangement used in 4-point measurement (distance between microphons: 5cm)

sound sources.

For convenience, the actual sound source is not distinguished from image sound sources. In addition, any image sound sources generated when there are two or more reflections are also treated as image sound sources. Generally, sound in a hall is reflected many times before it reaches the measurement point, so image sound sources are distributed like many virtual images in a mirrored

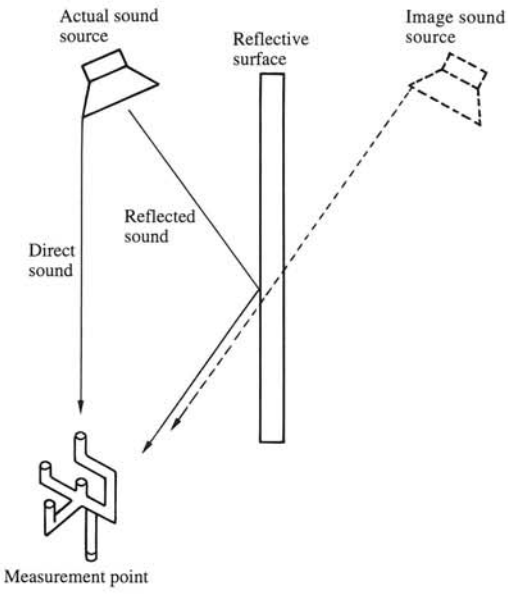


Figure 4. Virtual image sound source and reflection

room.

Measurements are made in a hall as follows: Impulse sound is emitted from a loudspeaker on the stage. Sound-field impulse responses are simultaneously recorded by the four microphones installed in seats (Figure 3).

Image sound sources are obtained from the four impulse responses. The details of how to obtain the image sound sources are explained in the next section.

The closely located 4-point microphone methed not only gives us the distribution of image sound sources. We can also plot the reflection energy distribution for each arrival direction, called a direction distribntion pattern (or hedgehog). It also makes possible analyses, such as analysis of reverberation time, D value (difinition), and frequency characteristics of the hall, using impulse responses.

4.2 Measurement of binaural impulse response using a dummy head

A dummy head is a model of the human head and usually has microphones at the locations of both eardrums.

By recording the output of the two microphones in a dummy head sitting on a seat in the hall, we can measure the approximative sound heard by a human listener.

Measurement of impulse responses of the dummy head microphones helps with various types of analysis. For example, binaural playback of convoluted music signals and the correlation coefficient between both ears can be calculated.

We measured impulse responses of the dummy head microphone using the apparatus shown in Figure 5.

4.3 Binaural recording of music recorded in an anechoic room

By convoluting the binaural impulse responses measured in Section 4.2 with music signals, the results become equivalent to those of binaural recording in the hall. However, this method has the drawback that convolution takes a long time because the impulse response time is long. To avoid this, we made a binaural recording. We played back music, e.g., a solo violin, flute, snare drum, or other instrument, recorded in an anechoic room from a speaker on the stage and recorded the output of the binaural microphones in the dummy head using digital audio tape.

4.4 Measurement conditions

We made measurements with no audience in the hall, at night, and with the air conditioners stopped to reduce background noise to a minimum.

Figure 5 is a block diagram of the measurement system. In measurement with closely-located 4-point microphone method, and measurement of binaural impulse response usiug a dumy head, we used not impulse sound but M-sequence signal. M-sequence signal is explained in the next section. We averaged the responses eight times to improve the S/N ratio.

We used a spherical enclosure with two speakers facing outward to make the sound source as omnidirec-

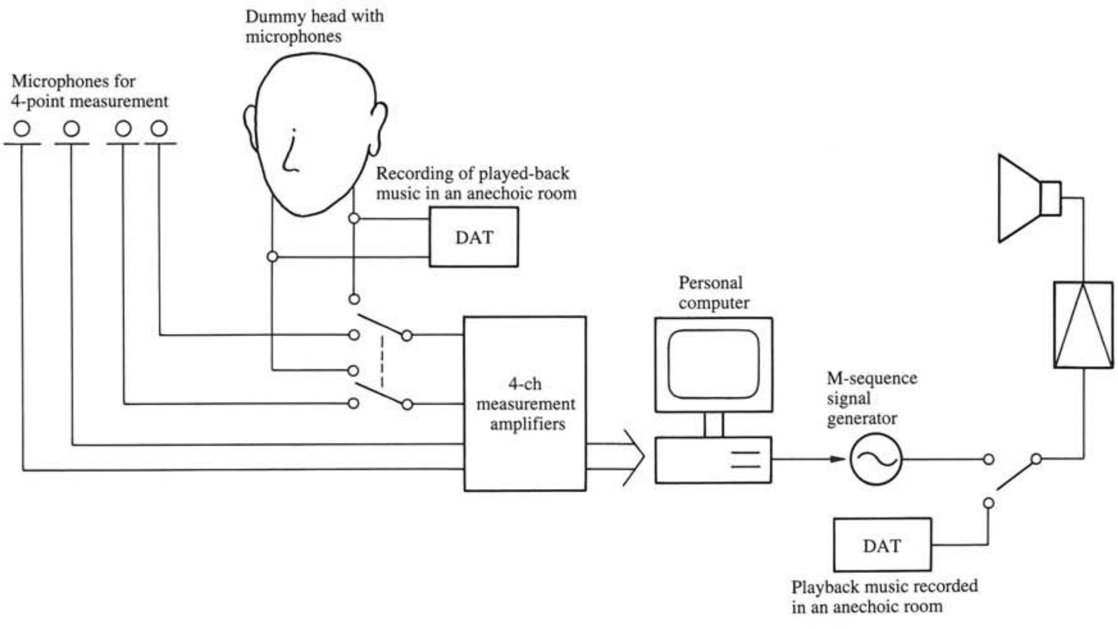


Figure 5. Block diagram of measurement system

tional as possible. We placed the sound source at 1.2m above the center of the stage.

We set up the 4-point microphones with the height of microphone O at the origin, 1.5 m from the floor. The Y axis was directed toward the front.

The dummy head was installed with both ears 1.5 m from the floor.

We placed three to seven measurement points in each hall. The measurement data in this paper is taken 12 m from the front of the sound source. If the measurement point was at a position where the correlation coefficient of both ears was 1, the position was shifted from the center of the seat. The measurement points are shown with a 1 in (b) in Figures 10 to 16 at the end of this paper.

5. Analysis

5.1 Measurement of impulse responses using M-sequence signal

Generally, to measure the impulse response of a hall, impulse sound is produced by a sound-source loudspeaker and the responses of the microphones are recorded. However, since the energy of single-pulse sound is smaller than that of uninterrupted sound such as white noise, averaging the responces (synchronous averaging) must be repeated many times to improve the S/N ratio. This means the measurements take a long time.

When we measured impulse response, we reduced the number of averaging responces using M-sequence signal to reduce the measurement time.

An M-sequence signal is an uninterrupted sound having flat, wideband frequency components and is frequently used as white noise. Figure 6 shows an example of an M-sequence signal waveform. The impulse response of a hall can be calculated by reproducing an M-sequence signal from the loudspeaker, and recording the M-sequence response of the hall, then calculating the correlation with the input M-sequence signal.

We measured four impulse responses in 4-point

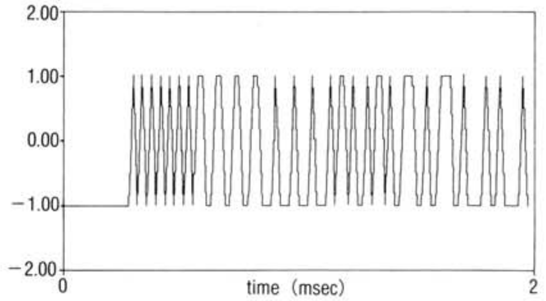


Figure 6. An example of an M-sequence signal waveform

method and binaural impulse responses of the dummy head at each measurement point. We used the responses for the analysis in Section 5.2 and subsequent sections.

We used an 18-order M-sequence signal to measure impulse responses for about 5.4 seconds with a sampling frequency of 50 kHz. During measurement, we averaged responses only eight times. However, we obtained a S/N ratio equivalent to or better than that obtained by making 128 times averaging using impulse sound.

5.2 Distribution of image sound sources

With the 4-point method, we end up with four impulse responses (Figure 7). At the beginning of an impulse response, the reflected sound arriving is weak and the reflection path lengths are not likely to coincide. Therefore, independently reflected sounds can be detected as independent pulses. A reflected sound reaches the microphones separated by some time lags.

For example, sound traveling in the Y-axis direction first reaches point Y. Sound arrivals at points O, X, and Z are simultaneous with a certain constant delay. In this way, the arrival time at each microphone can be uniquely determined.

The coordinates of a image sound source can be obtained from these differences in arrival times.

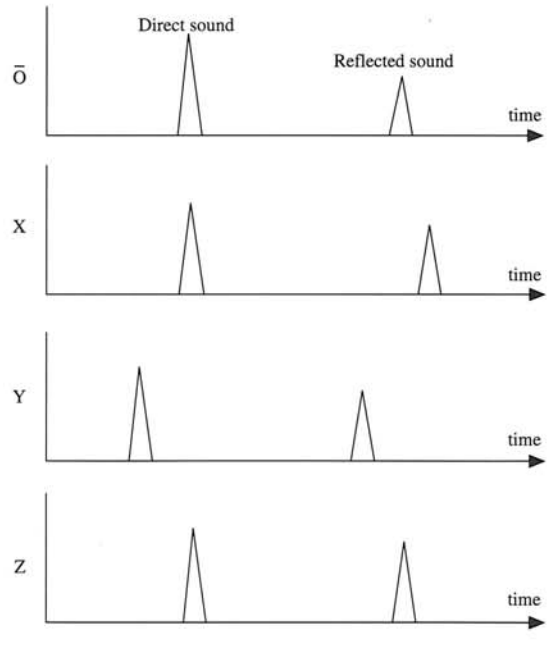


Figure 7. Four impulse responses

The energy of reflected sound in the impulse response at the origin (point O) is calculated and treated as the intensity of the image sound source.

In the distribution diagram, image sound sources are identified by circles. The center of each circle indicates the coordinates and the radius the intensity.

5.3 Direction distribution pattern (hedgehog)

Suppose that there is a virtual microphone with directivity (with a horizontal angle of one degree and a vertical angle between +45° and -45° as shown in Fig. 8). This microphone was used for a horizontal 360-degree scan. The sum of the intensity values of sound emitted from all image sound sources and entering the microphone was obtained for each horizontal angle and the direction distribution pattern calculated.

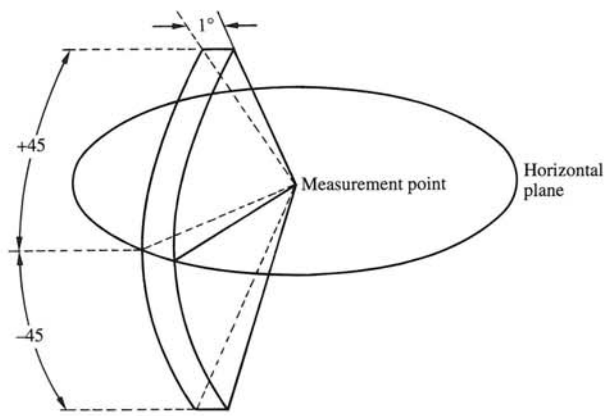


Figure 8. Directivity of virtual microphone used to compute the directional hedgehog

5.4 Reverberation time and D (definition) value

The reverberation times and D (Definition) values were calculated from the impulse responses measured using the 4-point method.

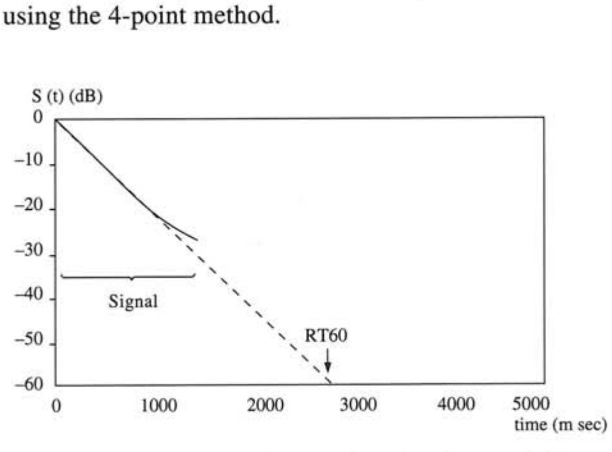


Figure 9. An example of a reverberation characteristic (dashed line is regression line)

The reverberation times were calculated as follows: First, a reverberation characteristic curve like the one in Fig. 9 was obtained from the impulse response. It was passed through a octave band filter according to the Schroeder equation (1) given below.

The reverberation time s(t) is:

$$s(t)\ (dB) = 10\ log_{10} \frac{\int_t^{\infty} P^2(\tau)d\tau}{\int_0^{\infty} P^2(\tau)d\tau} \qquad (1)$$

The time for the sound level to fall from 0 dB to -60 dB was obtained by applying the least squares approximation to the straight part of the characteristic curve.

The D values were calculated using equation (2). These values are related to sound articulation because they represent the size of the effect of the initial reflected sound.

$$\text{D values } (\%) = \frac{\int_0^{50m\,sec} P^2(t)dt}{\int_0^{\infty} P^2(t)dt} \times 100 \qquad (2)$$

6. Results of measurements

Figures 10 to 16 show the results of measurements. Results for one hall are summarized on one page. An explanation of each figure (a) to (e) is given below.

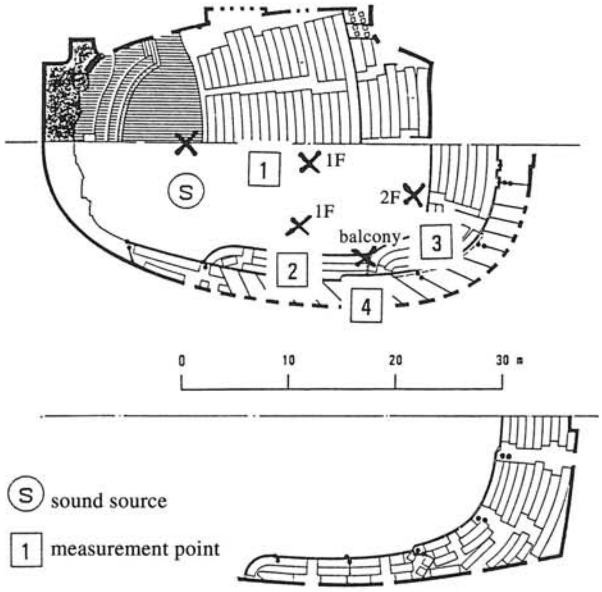
(a) Inside of the hall
A photograph taken looking out the stage of the hall. In the photograph of the Dom Zu Passau the photographer has his back to the pipe organ, i.e. the sound source.

(b) Plane and measurement points
The sound source and measurement points are marked with an 'X'. The sound source is marked with an 'S' and numbers 1 to 4 identify the measurement points. All of the results in this paper are measurements taken at point 1.
The volume, number of seats, and reverberation time are given below figure (b). The reverberation time was taken from the impulse responses passed through a octave band filter with a center frequency of 500 Hz (see Section 5.4).

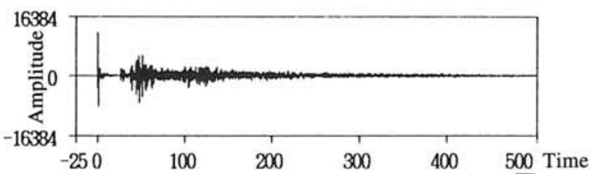
(c) Impulse response waveform
The unit of time (horizontal axis) is ms. The time when the direct sound arrives at the microphone is considered as 0 ms.

(d) Direction distribution pattern
Directivity is plotted from -50 dB to 0 dB assuming the maximum energy level in all directions is 0 dB. The upper figure is a top view and the lower figure is

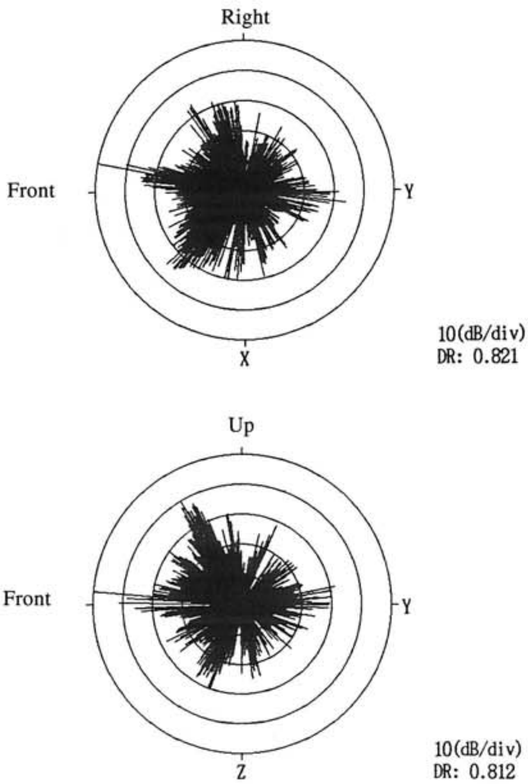
(a) Inside the hall



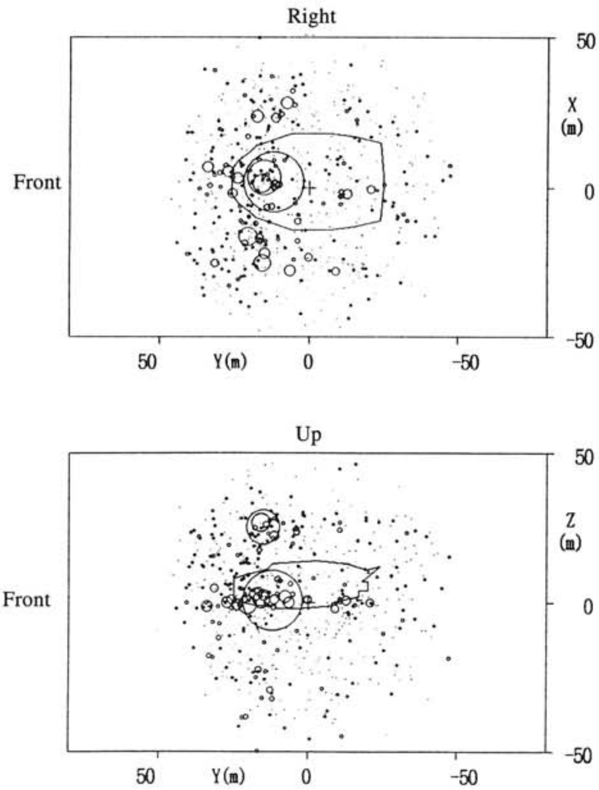
(b) Plan and measurement points
Volume: 12,500m³
Seats: 2,150
Reverberation time at 500Hz: 1.55sec



(c) Impulse response waveform



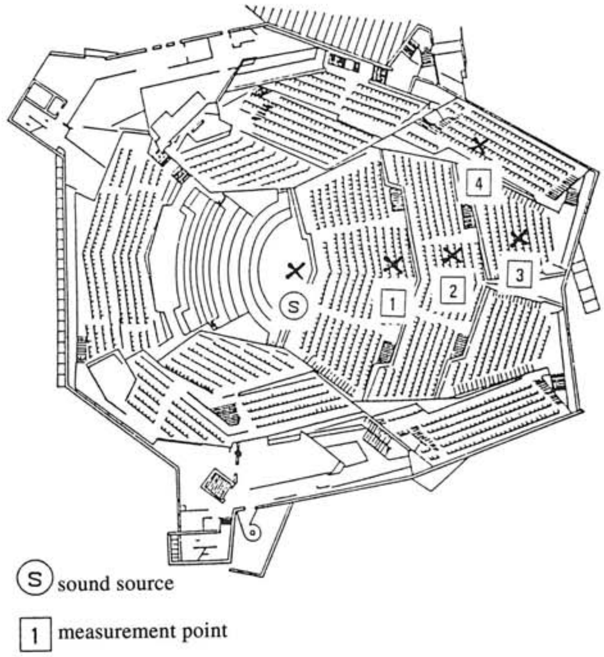
(d) Directivity diagram
(Upper: horizontal, Lower: vertical)



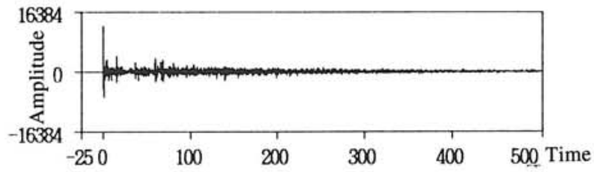
(e) Distribution of virtual sound sources
(Upper: horizontal, Lower: vertical)

Figure 10. Palais Des Beaux-Arts, Brussels, Belgium

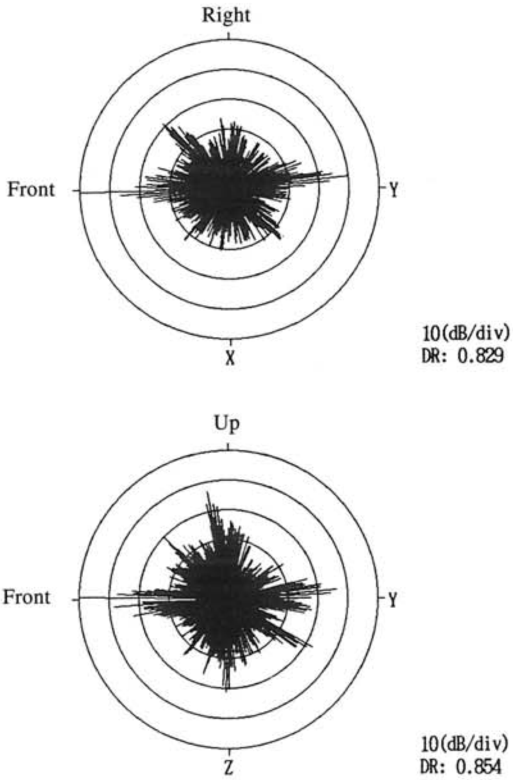
(a) Inside the hall



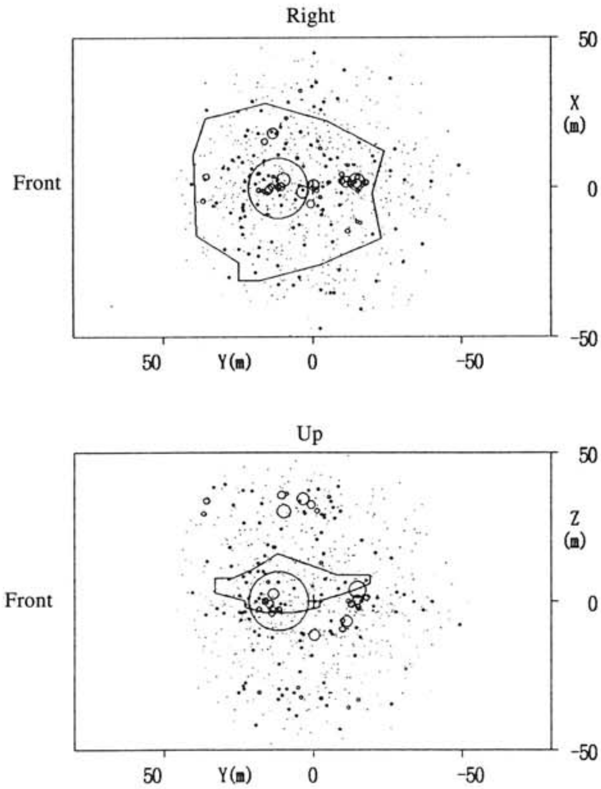
(b) Plan and measurement points
Volume: 26,000m³
Seats: 2,415
Reverberation time at 500Hz: 2.07sec



(c) Impulse response waveform



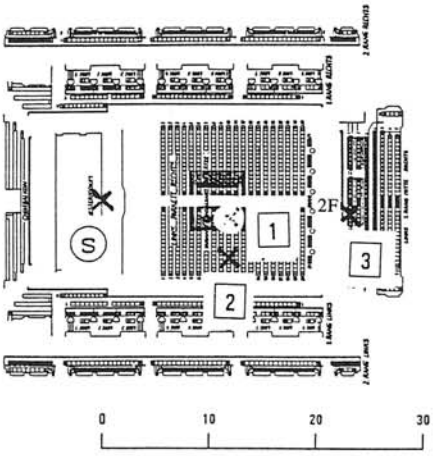
(d) Directivity diagram
(Upper: horizontal, Lower: vertical)



(e) Distribution of virtual sound sources
(Upper: horizontal, Lower: vertical)

Figure 11. New Philhormony Hall, Berlin Germany

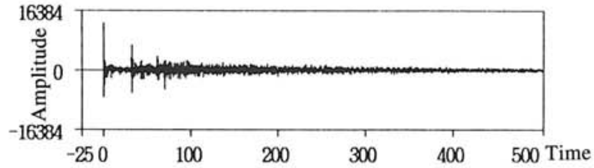
(a) Inside the hall



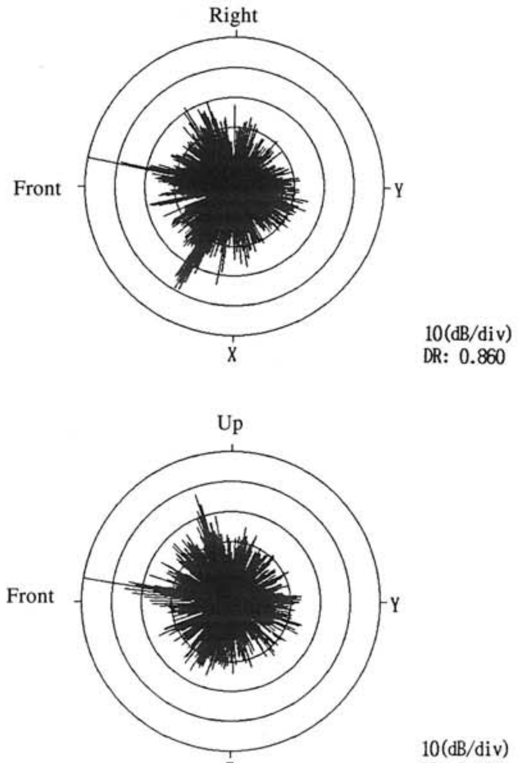
(S) sound source

[1] measurement point

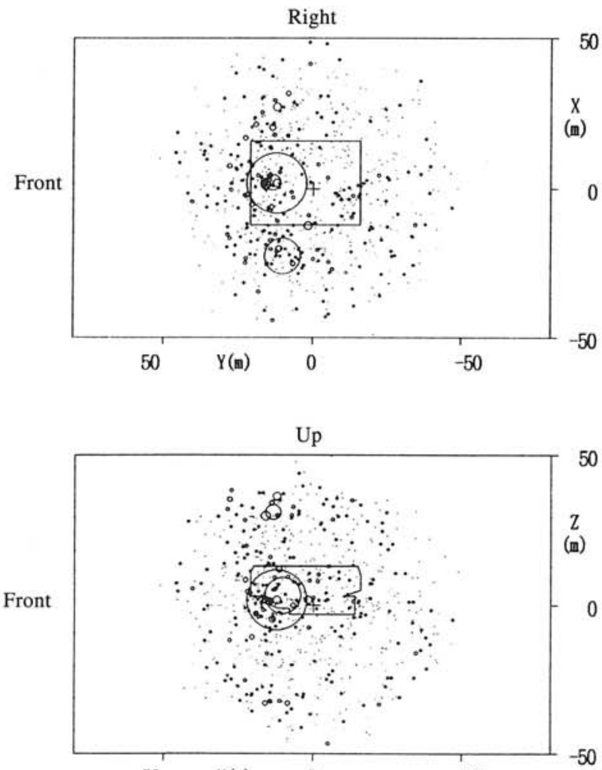
(b) Plan and measurement points
Volume: ——— m^3
Seats: 1,301
Reverberation time at 500Hz: 2.70sec



(c) Impulse response waveform



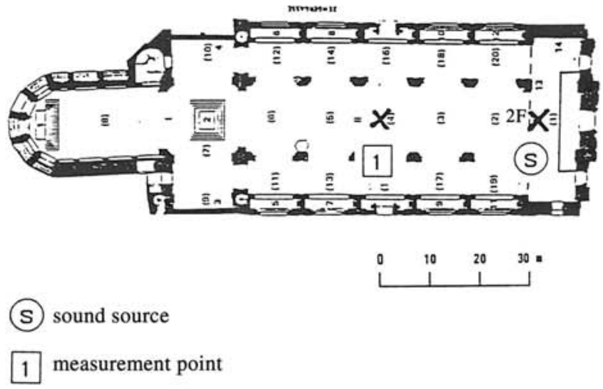
(d) Directivity diagram
(Upper: horizontal, Lower: vertical)



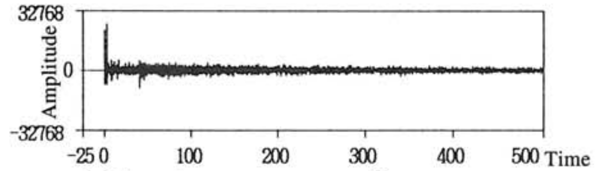
(e) Distribution of virtual sound sources
(Upper: horizontal, Lower: vertical)

Figure 12. Schauspielhaus, Berlin, Geranany

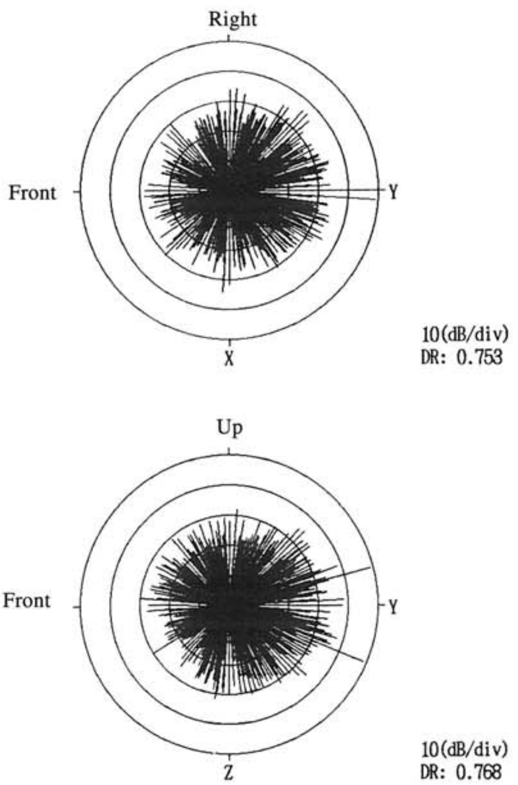
(a) Inside the Cathedral



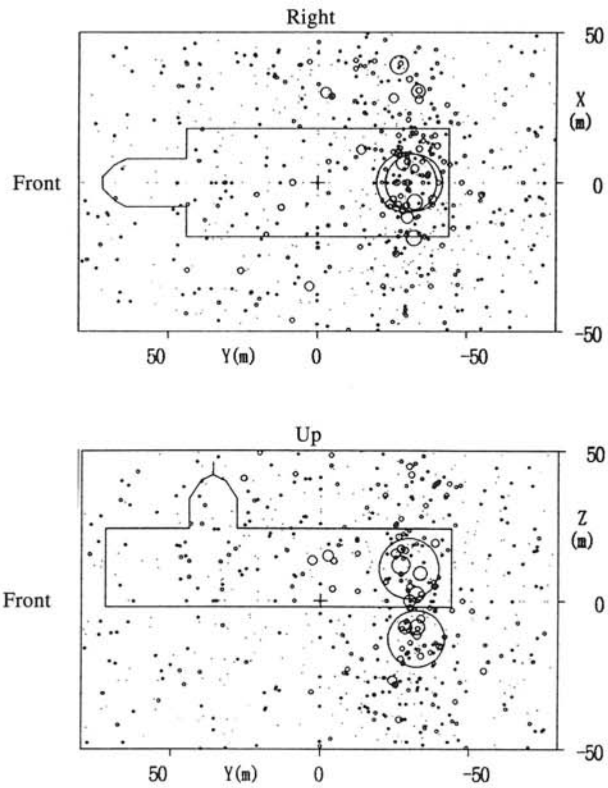
(b) Plan and measurement points
Volume: ———m^3
Seats: ———
Reverberation time at 500Hz: 4.96sec



(c) Impulse response waveform

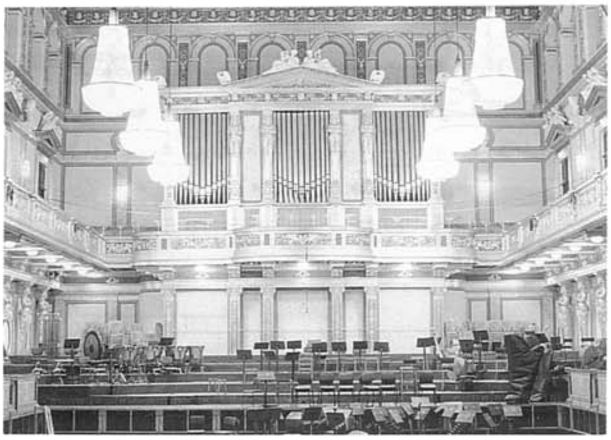


(d) Directivity diagram
(Upper: horizontal, Lower: vertical)

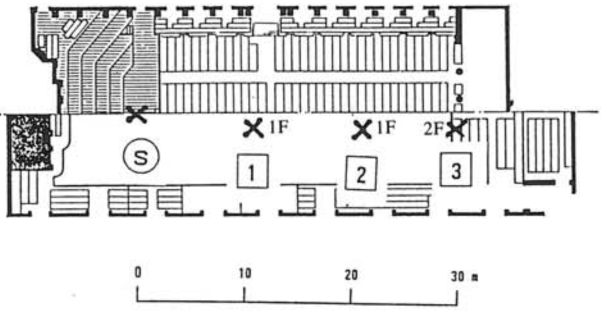


(e) Distribution of virtual sound sources
(Upper: horizontal, Lower: vertical)

Figure 13. Dom Zu Passau, Passau, Germany



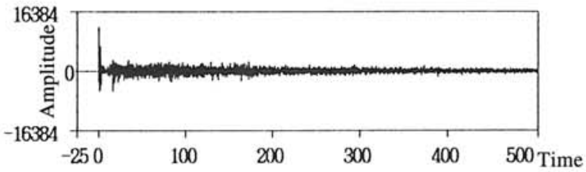
(a) Inside the hall



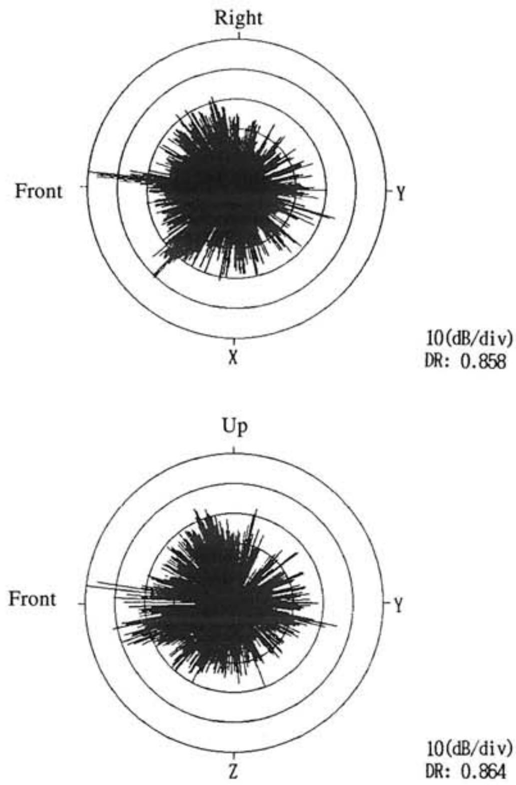
Ⓢ sound source

1 measurement point

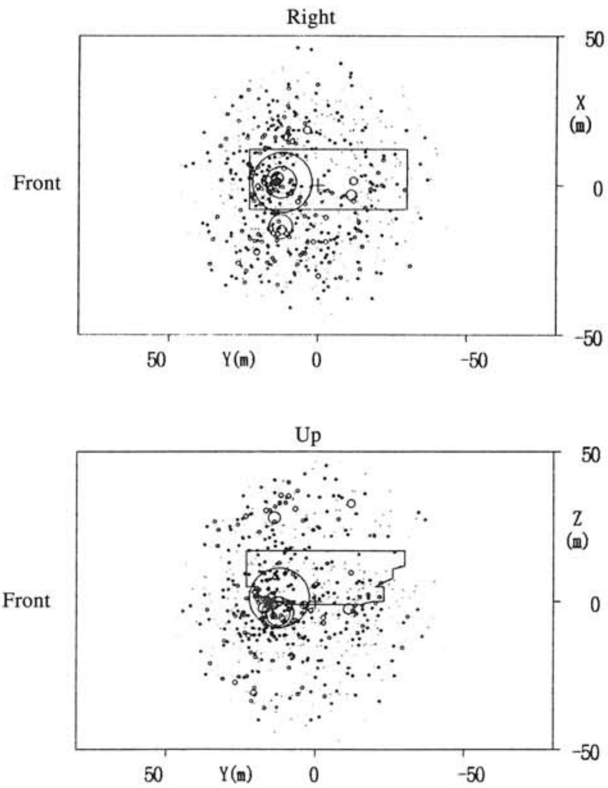
(b) Plan and measurement points
Volume: 15,000 m^3
Seats: 1,680
Reverberation time at 500Hz: 3.27sec



(c) Impulse response waveform

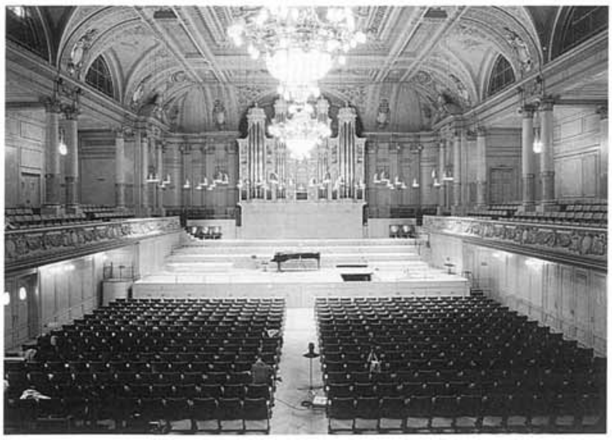


(d) Directivity diagram
(Upper: horizontal, Lower: vertical)

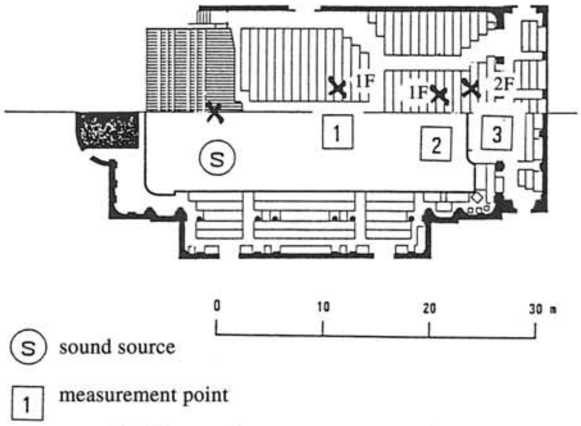


(e) Distribution of virtual sound sources
(Upper: horizontal, Lower: vertical)

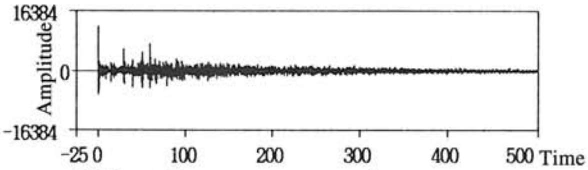
Figure 14. Grosser Musikvereinssaal, Vienna, Austria



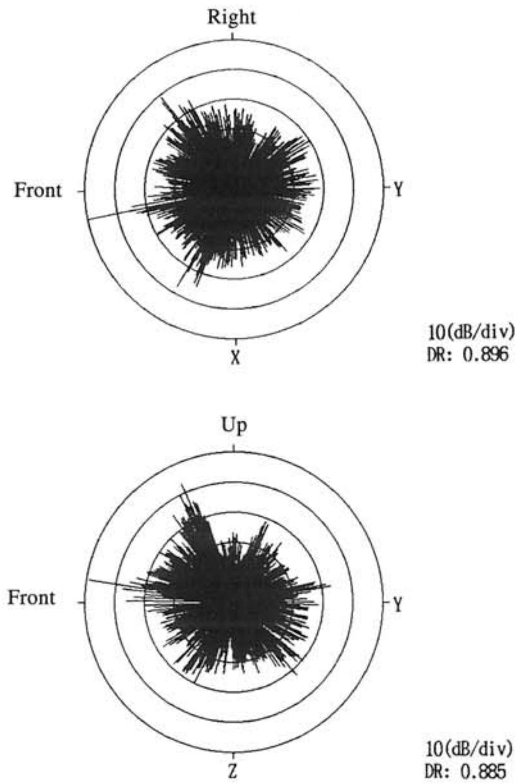
(a) Inside the hall



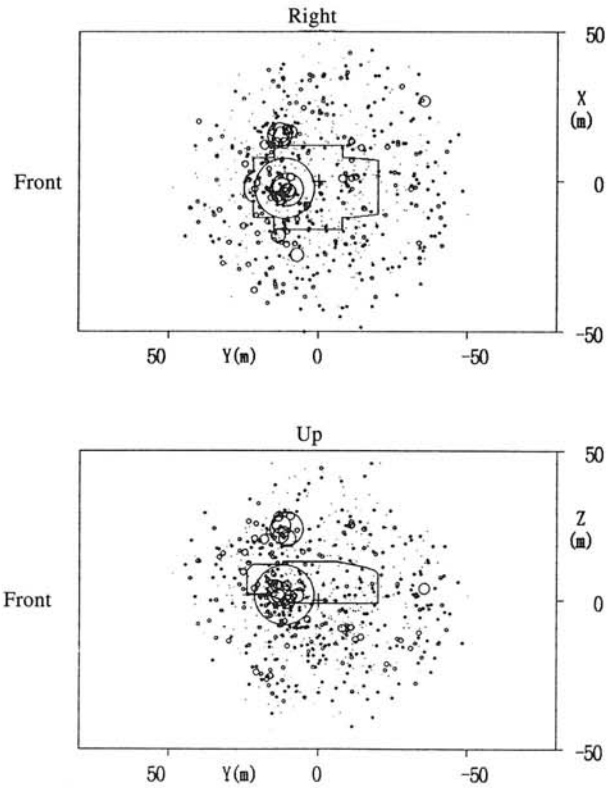
(b) Plan and measurement points
Volume: 11,400 m³
Seats: 1,546
Reverberation time at 500Hz: 3.59sec



(c) Impulse response waveform



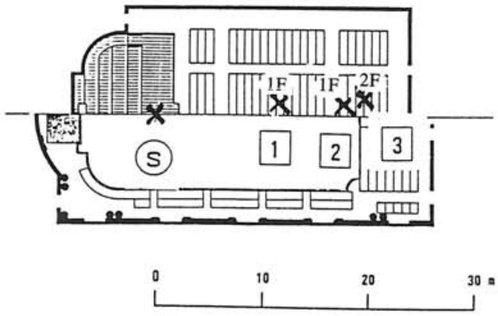
(d) Directivity diagram
(Upper: horizontal, Lower: vertical)



(e) Distribution of virtual sound sources
(Upper: horizontal, Lower: vertical)

Figure 15. Grosser Tonhallesaal, Zurich, Switzuland

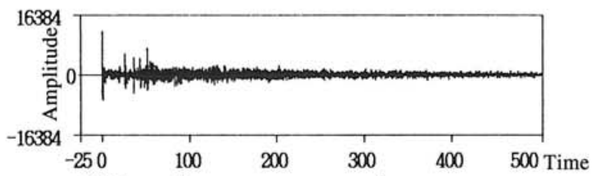
(a) Inside the hall



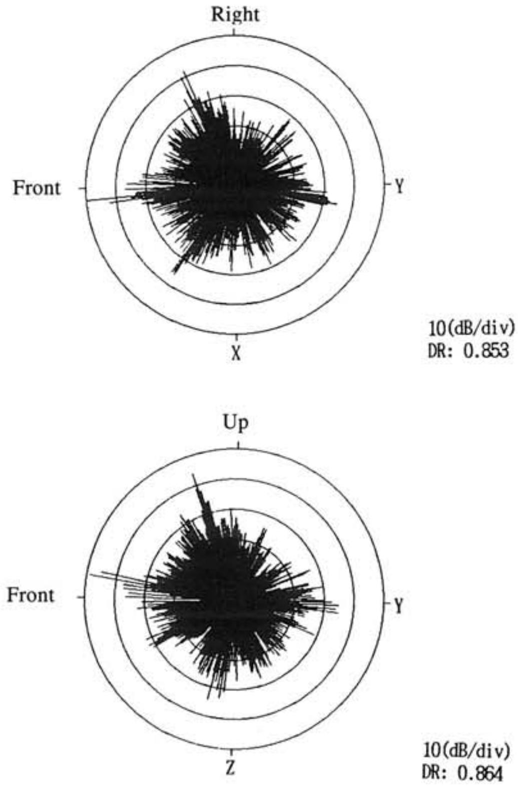
(S) sound source

[1] measurement point

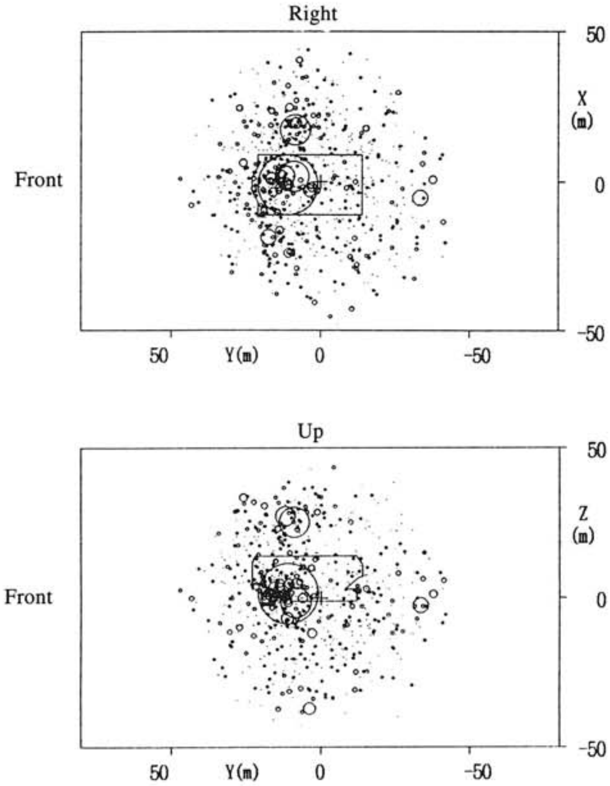
(b) Plan and measurement points
Volume: 10,500 m^3
Seats: 1,400
Reverberation time at 500Hz: 2.02sec



(c) Impulse response waveform



(d) Directivity diagram
(Upper: horizontal, Lower: vertical)



(e) Distribution of virtual sound sources
(Upper: horizontal, Lower: vertical)

Figure 16. Stadt-Casino, Basel, Switzerlaud

a side view. In both figures, the front of the hall is to the left.

The value DR in the lower right corner of each figure is the directivity diffusion ratio from 0 to 1. The closer the ratio is to 1, the more uniform the intensity of reflected sound from all directions.

(e) Distribution of image sound sources

Image sound sources whose intensity is from -50 dB to 0 dB are plotted assuming that the maximum image sound source level is 0 dB. The upper figure is a top view and the lower figure is a side view. In both figures, the front of the hall is to the left.

7. Conclusions

Most of the halls we visited are known for their good acoustic characteristics and have been subjects of measurements in the past. Some of these measurements have been published. Published figures are, however, not detailed enough for calculations of impulse response waveforms and reverberation time. The acoustic data available for all four shoebox type halls above are general characteristics of shoebox type halls, giving, for example, the direction from which sound reflected from the side walls arrives. When we heard sound in each hall, we found that each hall has own timbre and acoustic characteristics, something the published data could not tell us. To reproduce the individual sound of a hall, we had to collect our own data. For this reason, the measurements we made are an important contribution, they are also essential to improving our audio products. We have incorporated this data into one DSP chip to try to reproduce a natural sound and the characteristics of a particular hall and a particular position in that hall. We have had some success with our attempts. We also plan to use our data for overall product quality improvement, including engineer training. We can play the music we recorded via a dummy head to familiarize our engineers with the acoustic characteristics of famous concert halls. This kind of engineer training should improve the quality of our product designs.

8. Future plans

We plan to further develop our techniques for reproducing a natural sound in a passenger compartment by analyzing the data we obtained from these measurements and investigating the relationship between our data and human perception. If we have the opportunity, we want to measure and analyze the acoustic characteristics of halls, opera houses, jazz clubs, studios and other environments which differ from the halls we visited this time. We aim to increase the realism of our sound fields and create products which are truly attractive to users. We also plan to soon develop a new generation of products which automatically reproduce sound fields ideal for music played back in a passenger compartment.

Finally, we would like to thank assistant Professor M. Morimoto and Mr. K. Sakagami of Kobe University for their help and guidance. We would also like to thank the Ruhr University staff for their full support during our on-site measurements.

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